A Novel Buffer Underflow Avoidance Scheme for Multiple-source High Quality Multimedia Delivery

Seung-Bum Lee, *Student Member, IEEE,* Alan F. Smeaton, *Member, IEEE,*
and Gabriel-Miro Muntean, *Member, IEEE,*

***Abstract*—This paper proposes a novel buffer underflow avoidance scheme for multiple-source multimedia delivery which enables maintaining high user perceived quality in highly loaded network conditions. Unlike existing solutions which perform delivery adaptation by adjusting the original multimedia quality to varying network conditions, this solution is based on dynamic buffer occupancy estimation for multiple streams to achieve its goal. The proposed scheme and three other approaches are compared in terms of estimated user perceived quality. Simulation results show how this scheme outperforms the other solutions including when the number of simultaneous receivers increases significantly.**

***Index Terms*—Multiple-source streaming, buffered multimedia delivery, user perceived quality.**

I. Introduction

Compared to single source streaming [1], multiple sender-based approaches show good performance when dealing with variable network conditions during multimedia streaming. Some of these multiple source approaches make use of multicast overlay [2]–[4]. A UDP-based approach, PROMISE [3] introduces CollectCast as a multi-path live streaming which enables the best sender set selection in current conditions. Unfortunately these schemes use Constant Bit Rate (CBR) only for video streaming [5] and do not support real-life high quality video delivery which is in general Variable Bit Rate (VBR). In addition, they do not maintain high Quality of Service (QoS) levels.

There are several approaches that use unicast-based multiple source streaming [6], [7]. Among these, Nguyen and Zakhor propose Multiple Sender Distributed Video Streaming (MS-DVS), a framework for streaming video simultaneously from multiple mirror sites to single receivers over the Internet [7]. In order to increase tolerance against loss and delay due to network congestion, they adopt a rate allocation and a packet partition algorithm. However, they do not consider quality-related issues either.

The most significant problem in multimedia streaming is that the available network bandwidth does not match media encoding/sending rate. There are two major solutions to this issue: media adaptation [8] and buffer management [9], [10]. Media adaptation approaches adjust multimedia streams to the available network resources varying multimedia bitrate and therefore quality. The approaches using data buffering provide more flexibility to applications, but streaming quality is often affected by buffer under/overflow or delays. In general good buffer management determines very good user perceived quality.

S.-B. Lee and G.-M. Muntean are with School of Electronic Engineering, Dublin City University, Ireland. Email: {sblee, munteang}@eeng.dcu.ie, Website: http://www.eeng.dcu.ie/∼sblee and http://www.eeng.dcu.ie/∼munteang. A.F. Smeaton is with the CLARITY CSET, Dublin City University, Ireland. The support from Samsung Electronics Co. Ltd. and Science Foundation Ireland (grant 07/CE/I1147) is gratefully acknowledged.

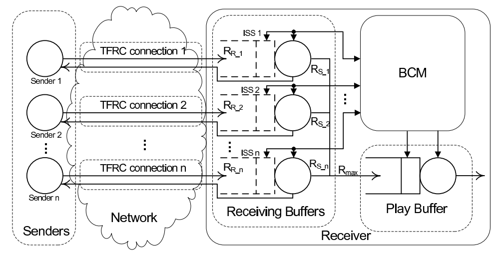

Fig. 1. Example of a QAMMD-based Multimedia Delivery System

This paper proposes a novel **Buffer Underflow Avoidance Scheme (BUAS)** for a Quality-oriented Algorithm for Multiple-source Multimedia Delivery (QAMMD), which maintains high quality of delivery without media quality adaptation to network conditions. In order to overcome varying network conditions, QAMMD employs a double buffering architecture which uses virtual multiple buffers associated with multiple network connections powered by BUAS in conjunction with the classic decoding/playing buffer.

The next sections describe QAMMD and BUAS in details and present simulation-based testing results. These results show how better quality of delivery is achieved by BUAS-based QAMMD in comparison with three existing solutions when they deliver VBR multimedia streams across an increasingly loaded network.

II. Quality-oriented Algorithm for Multiple-source Multimedia Delivery (QAMMD)

A. Double Buffering Architecture

QAMMD is an unicast-based multiple-source streaming approach. In order to support high quality multimedia stream delivery, QAMMD adopts a novel double buffering architecture. It includes n senders, one receiver and n network connections (associating senders to the receiver). Two levels of buffers are deployed at the receiver as shown in Fig. 1. These buffers include **multiple Virtual Receiving Buffers** and **a Play Buffer**. A novel **Buffer Coordination Module (BCM)** balances the functionality of this double buffering structure.

The multiple virtual receiving buffers are managed as **Individual Storage Spaces** (ISS). Each ISS stores multimedia data received via one of the n connections established between the multiple sources and the receiver. Although other protocols can be used for this purpose, QAMMD makes use of the TCP Friendly Rate Control (TFRC) protocol [11] in order to best balance the aggressiveness of the multimedia delivery with friendliness towards other traffic. Each ISSi receives data via the network with a rate R_{R_i} and provides data to the play buffer at a rate R_{S_i}. R_{R_i} is estimated using TFRC throughput [11], whereas R_{S_i} is determined based on dividing the maximum media encoding rate R_{max} by the number of senders. ISSs do not really store the data (the play buffer stores it for efficiency), but they enable BCM to control the data flow for buffer underflow avoidance.

The **Play Buffer** uses the MPEG Video Buffering Verifier (VBV) mechanism [12] with an unbounded buffer size. When the number of packets in the play buffer reaches the initial number of packets set for efficient buffering (S_{init}), the data is fed to the decoder and then to the player. This MPEG VBV operation guarantees that encoding-related factors do not cause buffer underflow in the play buffer given certain VBV buffer sizes, VBV delay and maximum media encoding rate R_{max}, as required by local playback [13]. In these conditions, the play buffer will receive data at the R_{max} rate, which can be determined at encoding time. Consequently, R_{max} is used as the aggregated target value for the overall ISS sending rate which is $\sum_{i=1}^{n} R_{S\_i}$. However, when setting S_{init}, current network conditions are considered.

The **Buffer Coordination Module** (BCM) controls data flow between the TFRC connections, ISS and play buffer. BCM involves packet partition and rate allocation mechanisms. BCM retrieves media related information such as VBV buffer size, VBV delay, and media rate (R_{max}) and manipulates buffer parameters based on the information from the buffers and multimedia data. In addition, it determines receiving buffer parameters such as R_{S_i}, S_{init}, etc. In doing so, BCM uses an innovative **Buffer Underflow Avoidance Scheme (BUAS)** which is described in the next subsection.

B. Buffer Underflow Avoidance Scheme

The proposed **Buffer Underflow Avoidance Scheme (BUAS)** considers the initial play buffer size, S_{init} as the maximum value between the initial VBV buffer size, (S_{VBV}) and estimated play buffer size, (S_{ab}). S_{VBV} is calculated at encoding time [12]: for the case of VBR, S_{VBV} is given; for CBR, S_{VBV} can be determined by multiplying the VBV delay and average data rate. BUAS estimates S_{ab} using a buffer underflow probability (BUP) analytic model [9] for each ISS.

Eq. 1 presents the closed-form BUP formula, $\gamma_i(x_i)$ which determines the probability that the total duration of all buffer underflow events is greater than 0 sec with an initial buffer size, x_i. α and β are the inverse of the expected values of buffered packet changes in decreasing and increasing states of a receiving buffer, respectively. $F_{\hat{\theta}_n}(\hat{\theta}_{\text{media}})$ is the value of exponential random variable cumulative distribution function of $\hat{\theta}_n$ with the rate, R_{S_i}. These parameters can be calculated using round trip time (rtt_i), R_{R_i} and R_{S_i}.

$$\gamma_i(x_i) \approx \left(F_{\hat{\theta}_n}(\hat{\theta}_{\text{media}}) + (1 - F_{\hat{\theta}_n}(\hat{\theta}_{\text{media}})) \frac{1 - e^{-\beta}}{1 - e^{-\alpha}} \right) \cdot e^{-(\alpha-\beta)x_i}, \quad x_i > 0 \tag{1}$$

The inverse of $\gamma_i(x_i)$, $\gamma_i^{-1}(P_i)$ can be computed where P_i is buffer underflow probability of ISSi.

$$\gamma_i^{-1}(P_i) \approx \frac{log\left(\frac{P_i}{F_{\hat{\theta}_n}(\hat{\theta}_{\text{media}}) + (1 - F_{\hat{\theta}_n}(\hat{\theta}_{\text{media}})) \frac{1-e^{-\beta}}{1-e^{-\alpha}}}\right)}{\beta - \alpha} \tag{2}$$

BUAS considers that play buffer underflow occurs when all ISSs have reached underflow. Using this assumption, the overall BUP (P) is the product of BUPs of all n ISSs as presented in Eq. 3

$$P = \prod_{i=0}^{n} \gamma_i(x_i) \tag{3}$$

In order to achieve the given target BUP, (P_{target}) in the play buffer, BUAS estimates the initial buffer size as in Eq. 4. P_{target} is dependent on the number of users to be supported and consequently connections to be established. The higher the number of connections, the lower the probability of buffer underflow. For example, a good target value for expected 180 connections is 200, making P_{target} 0.005 (i.e. 1/200) as this allows for a margin of error.

$$S_{ab} = \sum_{i=0}^{n} \gamma_i^{-1}((P_{target})^{\frac{1}{n}}) \tag{4}$$

In summary, BUAS determines required buffer size S_{init} before providing data to the decoder/player based on S_{VBV} determined during encoding time and S_{ab} which is estimated periodically using target play buffer underflow probability (P_{target}), received data rate (R_{R_i}), data rate to the decoder (R_{S_i}) and inverse function of ISS BUP function $\gamma_i^{-1}(P_i)$.

III. Experimental Results

A. Simulation Models, Setup and Video Sequences

Modelling and simulations use Network Simulator version 2 (NS-2) [14] and employ models for QAMMD, Predictive Buffering Algorithm (PBA) [10], a MSDVS-like [7] multiple TFRC connections-based approach (mTFRC) and a PROMISE-like [3] UDP-based multiple streaming solution (mUDP). QAMMD and PBA adopt the buffer estimation algorithm but they use different solutions. PBA uses a statistical approach which assumes the connections are not correlated. Instead, mTFRC and mUDP do not use buffer prediction. mTFRC uses adaptive data delivery based on the TFRC protocol, and mUDP uses equalised bandwidth allocation at the start of the streaming instead of dynamic bandwidth allocation which is used by the other solutions. In all approaches, the receiver requests the same packets to be delivered from the multiple senders. In addition, all approaches adopt static peer selection and initially connect to three senders only.

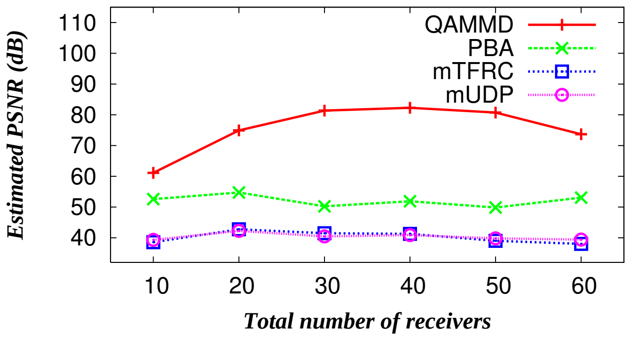

Fig. 2. Estimated PSNR with various solutions in uncongested network

In QAMMD, P_{target} is set to 0.005 and S_{VBV} is set to 224 kbytes which is determined at encoding time. The same S_{VBV} is used by mTFRC and mUPD, too. All approaches use 3.2 Mbps as target bandwidth which is higher than the encoding rate of 3 Mbps in order to cover network delivery overhead.

The simulation uses a "**dumbbell**" topology [6] with a 220 Mbps bandwidth and 5 ms delay bottleneck. All queues between links are drop-tail queues with a limit of 2000 packets.

Five five-minute long VBR encoded video sequences were selected from movies with different degrees of motion content: "Die Hard 1" - high, "Jurassic Park 3" - average, "Don't Say A Word" - average/low, "Family Man" - low and "Road To El Dorado" (cartoons) - average/high. The clips were MPEG-2 encoded at 3 Mbps using the same frame rate (25 frames/sec) and the same IBBP frame pattern (12 frames/GOP). Traces were collected from these clips and used during simulations.

B. Simulation Scenario and Result

The current test scenario randomly assigns a clip to each receiver from the set of clips mentioned above. The duration of simulation is set to 100 secs. All connections start at 1 sec after the simulation starts. The scenario includes 5 FTP connections as background traffic which start and end at the same time as the streaming connections. QAMMD, PBA, mTFRC and mUDP approaches are used in turn as multiple source streaming methods. The number of receivers is gradually increased in steps of 10. Since a receiver is assumed to have three senders, the total number of senders is three times as large as the number of receivers. For example, 10 receivers require 30 senders. Consequently, 40 nodes are involved in the 10 receivers test.

In uncongested conditions (between 10 and 60 receivers in the system), all approaches show no packet loss. However, loss with mUDP is significantly increased when congestion builds up. Relatively, TFRC-based approaches including QAMMD and mTFRC experience less loss, with QAMMD outperforming the other solutions at all times.

Fig. 2 shows a comparison between schemes in terms of quality, as estimated by Peak Signal Noise Ratio (PSNR) based on frame loss and throughput with increasing numbers of users. On average, when using QAMMD PSNR is 75.7 dB, whereas when PBA, mTFRC and mUDP are employed PSNR is 52.1 dB, 40.2 dB and 40.4 dB, respectively. It can be seen how QAMMD behaves with 45.3% better than PBA, with 88.3% better than mTFRC and with 87.4% better than mUDP. It can still be observed that the quality does not dramatically decrease for increased number of receivers. Specifically, when the bottleneck channel becomes crowded with 60 receivers, QAMMD offers 38.8% better perceived quality than PBA, 93.9% better perceived quality than mTFRC and 87.1% better than mUDP expressed in terms of PSNR.

IV. Conclusion and Further Work

This paper presents a novel **Buffer Underflow Avoidance Scheme (BUAS)** for a Quality-oriented Algorithm for Multiple-source Multimedia Delivery (QAMMD) which maintains high quality levels in highly loaded conditions during multimedia delivery. BUAS best balances the flow of data between the multiple connections enabling to achieve high quality while performing multimedia streaming without content adaptation to network conditions. Compared to other solutions making use of the Predictive Buffering Algorithm (PBA), multiple TFRC connections (mTFRC) and multiple UDP connections (mUDP), simulation results show that QAMMD obtains significantly better performance in terms of multimedia quality. Prototyping and subjective testing to complement these results are in progress.

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